VoIP over WiFi - degradation analysis with flat fading emulator

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***Abstract*— It is well known that the quality of real-time applications, like VoIP, in wireless networks suffers great degradation due to the signal instability in mobile environments. In this paper the quality of VoIP applications on WiFi networks is evaluated using an emulation system with variation of the test environment following the Rayleigh distribution. With this system it is possible to maintain control of the environment and modify it, emulating real situations of WiFi networks in obstructed places.**

***Index Terms*— Flat Fading, QoS, Rayleigh, VoIP, Weibull, WiFi.**

I. Introduction

Currently Voice over Internet Protocol (VoIP) and Wireless Local Area Networks (WLANs) are two technologies of great prominence in the market and in scientific research. The current status indicates that the convergence of these two technologies will create a new paradigm in the market of telecommunications and also in many Internet applications.

VoIP presents many advantages when compared to the traditional Public Switched Telephone Network (PSTN) mainly with regard to its cost, bandwidth efficiency and creation of differentiated services that can be offered to the users. The WLANs, in turn, offer the so desired mobility, easy installation and use and also low cost when compared to many wired network solutions. The low cost of VoIP allied to the mobility and popularity of the WLANs, more specifically WiFi, became VoIP over WiFi (VoWiFi) a promising application and the target of many studies, mainly regarding the performance of the network and the quality of service (QoS) required for the VoIP.

There are three great problems inherent to the WLANs that can harm VoWiFi performance: the inefficiency of the 802.11 MAC protocol, the signal instability caused by electromagnetic phenomena and the competition for bandwidth usage between voice traffic and data traffic. The article focuses on the analysis of the second problem: signal instability.

To test the quality of VoIP applications on 802.11b networks a wireless network emulation system was built in the Wireless Communication Laboratory at PUC-Campinas, in partnership with WCN-Intel Laboratory at UNICAMP. This project included the development of customized software that controls the emulation of flat fading channel with Rayleigh distribution.

II. WiFi

A. 802.11 standard

All WiFi equipments implement 802.11 protocol family, specified by IEEE. The standard 802.11a supports a maximum of 54 Mbps transfer rate, at 5.3 GHz. The standard 802.11b and 802.11g are more used than the previous one and support, respectively, a maximum of 11 Mbps and 54 Mbps transfer rate, both at 2.4 GHz.

WiFi networks use the CSMA/CA (Carrier Sense Multiple Access / Collision Avoidance) MAC (Medium Access Control) protocol, where the transmission collision is prevented. The stations access randomly the medium; they do not have any kind of privileges with regard to the service that is being transmitted [1]. Thus, real-time application as VoIP and video streaming suffer as much as Web applications and file transfer with network performance degradation. This scenario is not fair, because real-time applications do not work properly in networks with package loss, delays and jitter; on the other hand other applications (like e-mail) not even note these network problems.

Nowadays, there is a proposal which prioritizes some applications in the dispute for the medium; it is implemented by 802.11e standard, which specifies a new MAC protocol that can support until 8 different applications classes and offers different ways to guarantee the network quality [2]. This paper focuses on the analysis of 802.11b performance, because products that implement 802.11e are not yet available in the market.

B. Problems and Challenges

VoWiFi must surpass many challenges before reaching the quality offered by PSTN. The main challenges are: efficiency, range, end-to-end delay and jitter.

1) Efficiency: VoIP calls typically have low efficiency of useful data (payload) per network traffic (frames). When using high compression codecs, as GSM, the headers (data link + IP + UDP + RTP) represent more than 60% of the transmitted data. The total overhead caused by the headers (14 bytes of the data link layer, 20 bytes of the IP layer, 8 bytes of the transport layer and 12 bytes of the RTP) when compared with the size of GSM payloads, about 30 bytes, proves the low efficiency of VoIP in the use of the network bandwidth.

Besides that, the performance of wireless network (useful time slots during transmission) is much worse. To transmit a typical package of 85 bytes it is taken about 62 μs $(85 * 8/(11 * 10^6))$ at 11 Mbps. The total delay time imposed by the MAC and PHY layers of 802.11 (physical preamble, MAC header, MAC backoff time, MAC acknowledgment (ACK) and intertransmission times of packets and acknowledgment) represents about 800 μs, that is, almost 13 times the data transmission time [3]. The result of this is that the total efficiency of an 802.11b network does not reach 3% of the nominal throughput (11 Mbps).

This problem drastically reduces the maximum of simultaneous calls in a network. Using codecs such as GSM or G.729, theoretically it is possible to get only about 11 simultaneous calls [3], but this number varies according to the size of the wireless network, the concurrent traffic and the environment.

2) Range: In outdoor places with line of sight between all the stations and the AP a long range of the signal (about 70 meters) is possible, however, in indoor environments with an obstructed line of sight (NLOS environments) the attenuation of the signal significantly decrease the range of the network.

Mobile networks present a bigger limitation in its range, due to the negative effects that the movement of the stations cause. The signal quality of a mobile station near its AP can be worse than the quality of a fixed station further away, for example. In mobile environments an excellent site survey is essential for the good performance of the network. It is essential both, the correct positioning and sizing of the APs, and a detailed analysis of the network requirements for the applications that will be used.

3) End-to-end delay: The difference between the instant that the transmitter sends the first bit and the instant that the receiver receives it is called end-to-end delay. A very significant increase in this parameter can harm a real-time communication, because one speaker can overlap the other if there is a great delay between the first speaking and the second listening.

The components of the end-to-end delay are: propagation delay, encapsulation delay, delay in stations of the network and delay due to buffering.

4) Jitter: Jitter is the variation of the delay that packages take to arrive at the receiver. This difference occurs, for example, because the packages can pass through distinct ways in the network or wireless MAC changes. In real-time applications, like VoIP, this is the parameter of great significance in the measurement of the quality of the service. The presence of jitter in VoIP communication causes imperfections in the voice, which cannot be recovered.

III. VoIP

The VoIP technology allows the codification, digitalization and encapsulation of voice in IP packages. In the transport layer, the VoIP applications use the UDP protocol (User Datagram Protocol). In the application layer typically it is used the RTP/RTCP protocols, wich was developed for this purpose (real-time applications). This type of application requires high interactivity and very low reply time between the parts that are communicating. Due to these facts, to validate a VoIP application it is necessary a methodology that evaluates the quality of the service in one determined network. The most accepted criterion to measure the quality of a VoIP call is the Mean Opinion Score (MOS) [4], which consists of the arithmetic average of marks (from 1 - bad to 5 - excellent) attributed by listeners to the recorded tests phrases. The Table I shows the MOS scale and the quality of each mark. The target of any communication system is to get a MOS greater or equal to 4; however, in the common telephony this value can be lower than 3.7 and 3.2 in cellular networks [12].

TABLE I
MOS SCALE - QUALITY OF VOIP

Value	Score
1	Bad
2	Poor
3	Fair
4	Good
5	Excellent

The parameters that cause the major interference in the voice quality in VoIP communications are: end-to-end delay, delay variation of the arrived packages (jitter) and package loss. These parameters and its thresholds can be observed in Table II.

TABLE II
VOIP QUALITY PARAMETERS

Quality	End-to-end delay (ms)	Package loss (%)	Jitter (ms)
Excellent	< 150	0	0
Good	< 250	3	75
Fair	< 350	15	125
Poor	< 450	25	225

The perceptive evaluation to calculate the MOS is expensive and difficult to be made; other specifications had been created to evaluate these parameters computationally. Currently there are two recomendations from ITU-T that evaluate speech quality as an objective method. The first standard is the G.107 recommendation [7] that describes the E-model, which evaluates the degradation effect of the components of a telephonic conversation. The E-model results in a value called

R-factor (that varies of 0 to 100), that has a correspondent MOS value, as shown in Figure 1. The second standard is the P.862 recommendation [5] that results a value called of PESQ-MOS. It is directly related to the MOS value from P.800 recommendation, but ranges only from 1.0 (worst) up to 4.5 (best), diferently from the MOS that ranges up to 5.0. The explanation is: PESQ simulates a listening test and is optimized to reproduce the average result of all listeners. Statistics however prove that the best average result that can be generally expected from a listening test is not 5.0, instead it is 4.5.

Fig. 1. Relation between R-Factor and MOS score

A. Codecs

The transmission of voice in the Internet is in digital form, in other words, it needs to be codified. This process of sampling the analogical voice in digital information is called codification-decoding (codec). Codecs can be classified in two classes: based on the waveform or parametric codification (Vocoders) [12]. This paper concentrates on evaluating the quality of the VoWiFi using only codecs of the first class. In this family the codecs use distinct techniques for the codification, prioritization of the quality of voice, reduction of the amount of packages sent to reduce the bandwidth usage, suppression of silence and cancellation of echo, which results in different performance of them.

It was used in the tests only two codecs: G.711 (α-law) and GSM, because many of the other existing codecs are not cost-free and not implemented in the most popular softphones. This choice was made because the first codec (G.711 α-law) uses the PCM (Pulse Code Modulation) standard, which is also one of the most used standards; and the second one (GSM) is used in some cellular network and it uses a different codification standard: Regulate Pulse Excitation Long Term Predictor (RPE-LTP). In Table III it is shown the characteristics of each codec.

TABLE III
CODEC CARACTERISTICS

Codec	G.711a	GSM
Codification	PCM	RPE-LTP
Bit rate (kbps)	64	13
Sampling rate (kHz)	8	8
MOS	4.1	3.6

IV. Flat Fading

All mobile networks have problems with the variation of the signal intensity. The evaluation of the impact that these problems cause in the network is something too complex to be extracted experimentally due to probabilistic nature of all the electromagnetic phenomena that the signal suffers. In this article it is used a system of channel emulation that makes possible the reproduction of these phenomena in an 802.11 network with control and confidence to characterize the environment impairments. This platform emulates a wireless environment that can be characterized as a flat fading with Rayleigh distribution where it is possible to change the fading severity.

The flat fading phenomenon happens normally in environments with many obstacles, as in urban centers, where a path with line of sight between the sender and the receiver does not exist, and in indoor places, where there are many obstacles between the users. In this type of environment the signals suffer attenuation, refraction, diffraction and reflection in the path between the sender and the receiver, arriving with different delays and different components of amplitude and phase. The Rayleigh distribution is usually used when modeling the envelope variation of a signal when submitted to the flat fading [8] [9]. The flat fading observed in indoor environment has a Rayleigh distribution but with different severity.

In environments with no line of sight and with great attenuations and scatterings, the received signal can be represented as the addition of different transmitted signals that have suffered random attenuations during the multipaths.

In order to describe the multipath propagation of the transmitted wave, we can use the Rayleigh distribution as a particular case of the Weibull distribution, expression (1), with a shape factor α of two and a variable scale parameter β [11]. This Weibull(2,β) distribution is also called Rayleigh distribution with parameter β, denoted Rayleigh(β), and this factor is used to characterize the environment severity. In Figure 2 it is possible to observe the effect of β changes in the Rayleigh distribution. This flexibility is very useful to characterize the indoor changes and emulate the flat fading severity.

$$f(x) = \alpha\beta^{-\alpha}x^{\alpha-1}e^{-(x/\beta)^{\alpha}} \quad (1)$$

V. Methodology

For the tests an emulation system based on the Rayleigh distribution was used. This system has an isolated Access Point (AP) inside a shielded box, where the entire signal emitted by the AP is transmitted to a station through a coaxial cable. The transmitting signal is controlled by an RF (Radio Frequency) attenuator, responsible for modifying the receiving signal power. This attenuator, in turn, is controlled by a digital-to-analog converter and by another circuit that makes possible the adjustment of the offset and gain. This last circuit is

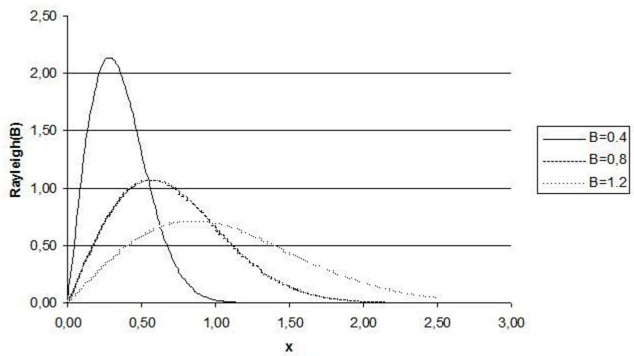

Fig. 2. Rayleigh Distribution

controlled by software, which was developed especially for these tests.

With this software, installed in a dedicated computer (control computer), it is possible to control the radio signal envelope, emulating a channel flat fading with Rayleigh distribution. In the other side of the coaxial cable that is connected to the AP there is another computer with an 802.11-PCMCIA card. The project of the emulation system is shown in Figure 3.

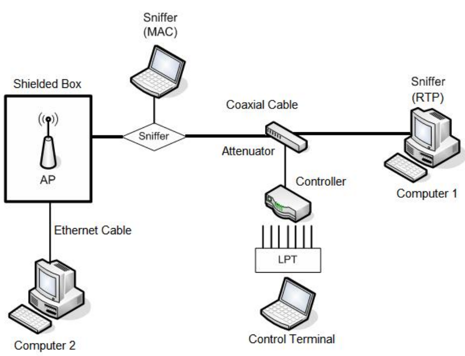

Fig. 3. Emulation system schematic

During the tests two computers were used to establish the VoIP calls. In the first computer there was an 802.11-PCMCIA card connected to AP through the coaxial cable described previously. The second computer was connected directly to AP through an Ethernet cable. In Figure 3 this configuration is shown (computer 1 is connected through the coaxial cable and computer 2 is connected through an Ethernet cable). For the data analysis it was used three commercial softwares: Wireshark, NetStumbler and Fluke Networks Optview. All the VoIP calls had been established using the software Xlite.

Computer 1 was responsible for capturing all RTP traffic of the conversation using WireShark and for monitoring the received signal power with NetStumbler. In computer 2, Optview VoIP Tool was used to analyze the effects of degradation in the call quality, through the measurement of the average delay, the jitter, packet loss and, finally, the MOS score and the R-factor. Figure 4 shows the emulation system.

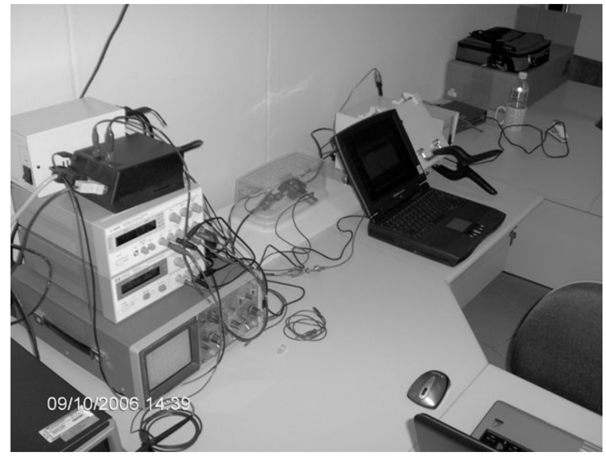

Fig. 4. Emulation system photo

The tests had been made with two codecs: G.711a and GSM. For each call established, each codec was chosen and the flat fading channel was emulated with Rayleigh distribution. Different values of β parameter of the Rayleigh distribution had been used, representing environments with different levels of degradation. For each test scenario five calls were established with different audio samples that simulate real conversations, generated as specified in ITU-T P.561 standard [6].

VI. Results

Based on the described methodology, we created different tests scenarios to perform a more elaborated data analysis. The flat fading effect, according to a Rayleigh distribution, emulates the movement of a person walking at a constant speed around an access point; therefore we keep the speed equivalent to a person walking (6 km/h) and modify only the β values of the Rayleigh distribution, thus considering environments with different levels of obstruction. We also consider that there was no competing traffic in the network, only the established conversation. In this way, for each β value, we turned on the flat fading effect and established a call using the software X-Lite. This way, it was possible to analyze the effect in the signal (using NetStumbler) and the quality of the voice (using the software OptiView Protocol Expert from Fluke Networks). We could see that when beta was increased the variation of the signal was increased too.

Figure 5 shows the signal in the receiver measured by NetStumbler; sample number 1 presents a very good signal, like a person stoped at a good position related to the AP; sample number 2 is the signal with the Flat Fading effect using β value equal 1.2, representing several obstacles; sample number 3 represents the average signal of the flat fading effect and sample number 4 is the signal used when simulating a standing person with bad signal.

When we increased the value of beta, to emulate a more obstructed environment, it was possible to identify a degradation in the quality of service (PESQ-MOS) shown by Fluke. According to the software manual [10], the values obtained are all in the same quality level, “Reach Connection“, which means that

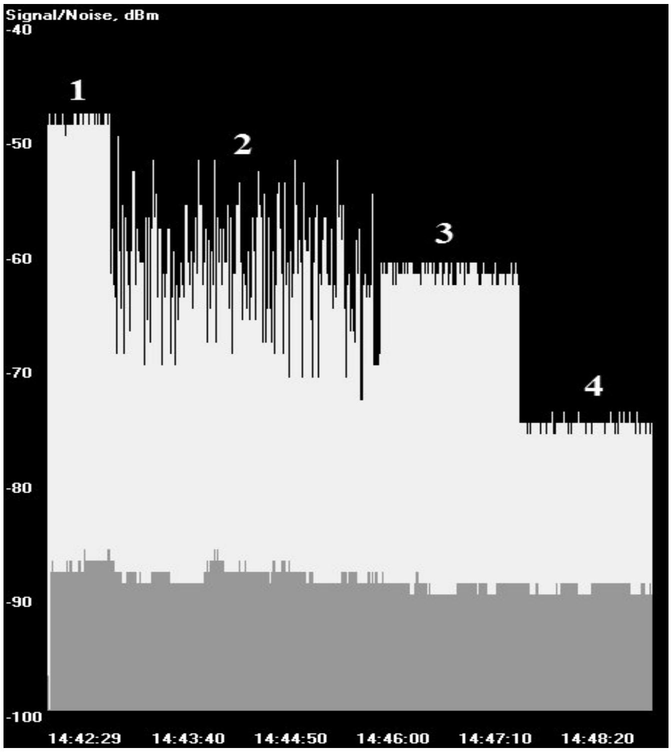

Fig. 5. Signal variation examples, using Netstumbler

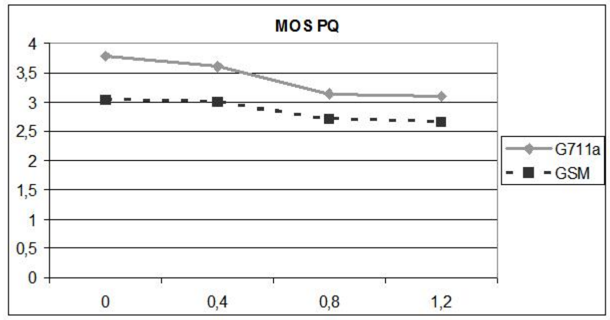

Fig. 6. PESQ-MOS decrease with different β values

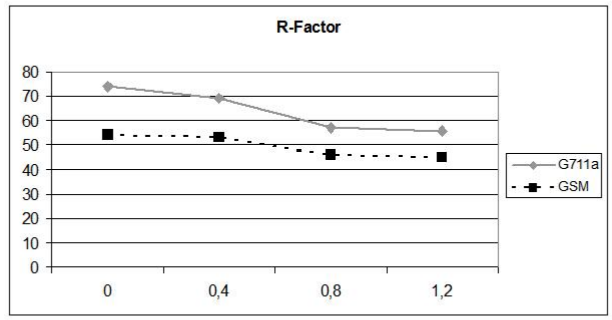

Fig. 7. R-factor decrease with different β values

the conversation quality was low but they still can understand each other. The obtained values were in the expected range, in other words, it had certain degradation in the quality when the beta increased. We believe that the degradation was not big enough because did not have competing network traffic or any other established connections; only a single call. During the tests all the established calls had been heard and evaluated subjectively. It was possible to notice that the conversations had suffered falls in quality, creating some failure moments in the communication and variations in the voice tone.

With the tests using different codecs, G.711a and GSM, it was possible to notice that GSM is the most robust codec although it does not supply high quality of service, therefore it obtains a lower variation in PESQ-MOS, even when the environment adversities increase. On the other hand, G.711a has a greater decrease in the quality, since it depends on a bigger bandwidth and on sending of a large amount of packages, so any variation in the signal implies in a great loss of packages. These results is shown in the Figure 6. Another good index to analyze is the R-fator shown in Figure 7. As we can see, both index vary equally, representing an objective evaluation of quality of service in VoIP applications.

For both codecs we made a comparative test without the flat fading effect, as if the person was standing still with worse average signal. The result is represented by the β equal zero in the graphs. With this we could analyze the interference in the signal caused by the movement. The most interesting thing is that the obtained MOS in these tests was near the one obtained when we considered a flat fading with a small β. This makes we believe that mobility causes great interference in the signal, and that, therefore is more interesting to be motionless with a bad signal, than in movement with a better signal.

Another good parameter to analyze is the jiter variation, which is shown in Figure 8. With the increase of β the jitter increases too, but in a different way for each codec. As we said before, GSM is more robust and can tolerate a huge signal variation. In this case when emulating a standing person the jitter was a little bigger when comparing with a small β, this happened because the average signal was worse.

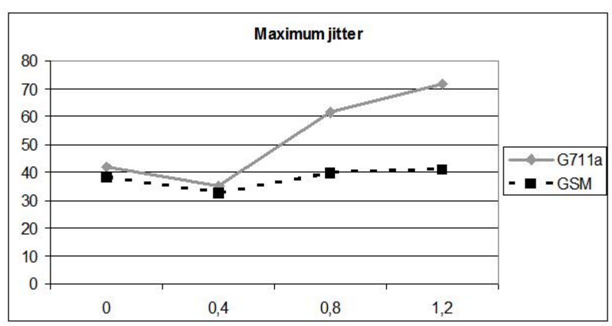

Fig. 8. Jitter increase with different β values

VII. Conclusion

In this work it was evaluated the effect of the signal variation, in indoor and obstructed environments, in VoIP applications. It was used an emulation platform with the implementation of the Rayleigh distribution that could be controlled to emulate different testing environments in an 802.11 network.

The flat fading phenomenon modified the signal substantially, what made the quality of the VoIP calls decrease significantly. It was possible to observe an increase in jitter and other parameters that influence in the quality of service of VoIP applications, such as the delay and the package loss. With these data it is possible to infer that it is better for a wireless station stay motionless at a weak signal level than in movement with high signal average.

Another interesting point is the fact that the low bandwidth codecs, like GSM, tolerate this phenomenon better. The differences in peformance when the calls with GSM codec were submitted to different flat fading environments were smaller than the differences observed in the calls with G.711a.

The future works will use this same platform to evaluate VoIP applications and will have as objective mainly 3 aspects: it will analyze the effects caused by concurrent UDP and TCP traffic in the network, as well as analyze the performance effects caused by the increase of simultaneous calls and, finally, the effects that stations with weak signal cause in stations with good signal quality in real-time applications.

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